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(54) Alternative telephone call routing system

(57) A telecommunication system includes a source private branch exchange (12) that transmits telephone calls from a source to a destination private branch exchange (26) over a public switched telephone network (18). As an alternative to transmitting the calls over the public switched telephone network, the private branch exchange is coupled to a telephony Internet server 30 that can transmit telephone calls over a global wide area computer network such as the Internet (32). The private branch exchange (12) queries the destination private branch exchange (26) to determine if it similarly coupled to a telephony Internet server. If so, the bandwidth used by the telephony Internet server will allow the transmission of the telephone call, the call is routed over the Internet (32) to the intended recipient.

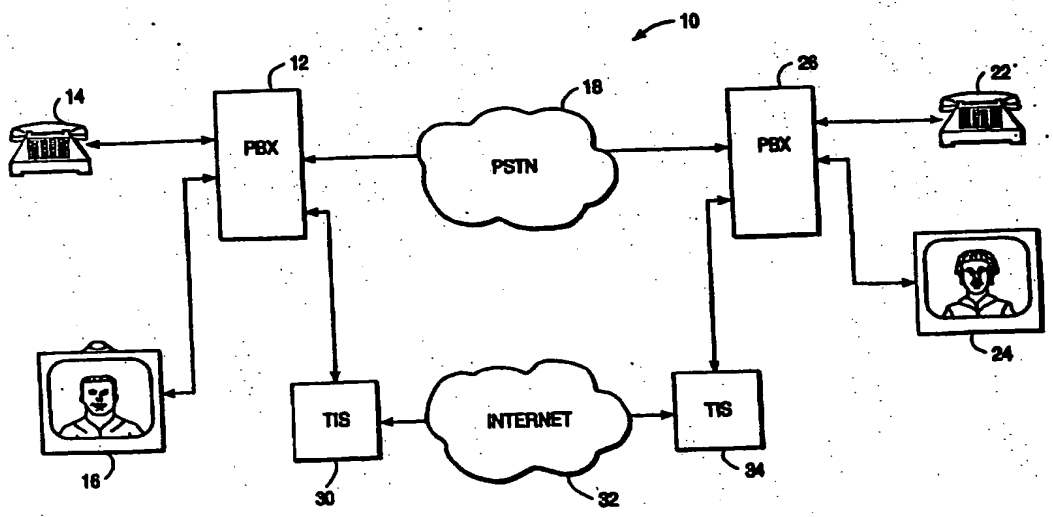

FIG_1

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Description

Field of the Invention

[0001] The present invention relates to communication systems, and in particular to systems for selecting alternative routes for telephone calls.

Background of the Invention

[0002] As business is conducted over ever expanding geographical areas, the use of telecommunication services to transmit voice and data signals is continually increasing. As a result of the increased use of these services, the cost for such services becomes a significant part of the cost of doing business. Therefore, most businesses are continually looking for ways to reduce their communication costs.

[0003] One known method of obtaining "free" telephone calls is to transmit the calls via a global wide area computer network such as the Internet. In effect, these calls are not free because the user leases the telephone lines that are used to provide their Internet access, however, calls placed over the Internet are not separately billed.

[0004] In the past, it has not been possible to seamlessly integrate the Internet as an alternative mute for transmitting telephone calls because there is no way of knowing whether an intended recipient has the ability to receive calls.

[0005] Given the shortcomings in the prior art, there is a need for a telephone communication system that can reduce communication costs by automatically determining when it is possible to transmit calls on the Internet to utilize excess bandwidth.

Summary of the Invention

[0006] To reduce the cost associated with transmitting telephone calls over a public switched telephone network (PSTN), the present invention is a communication system that transmits calls from a source private branch exchange or central office to a destination private branch exchange or central office. The source private branch exchange is coupled to a telephony Internet server that can transmit a call over a global wide area computer network such as the Internet. To determine whether the call can be transmitted over the Internet, the source private branch exchange transmits a message to the destination private branch exchange over the PSTN to determine whether it is similarly equipped with a telephony Internet server. If so, and the bandwidth available on the Internet will accommodate an additional call, then the telephone call is routed to the Internet.

[0007] The quality of the call placed on the Internet is continually monitored. If the quality drops below a predetermined threshold, the call is rerouted from the Internet back to the public switched telephone network.

Brief Description of the Drawings

[0008] The foregoing aspects and many of the attendant advantages of this invention will become more readily appreciated as the same becomes better understood by reference to the following detailed description, when taken in conjunction with the accompanying drawings, wherein:

FIGURE 1 is a block diagram of a communication system in accordance with the present invention; and

FIGURE 2 is a flow chart of the steps performed by the present invention to select the most appropriate path to route a telephone call.

Detailed Description of the Preferred Embodiment

[0009] The present invention is a communication system that can automatically determine whether to route a telephone call on a public switched telephone network or on a alternative path, such as the Internet, in order to reduce communication charges.

[0010] As shown in FIGURE 1, the communication system 10, according to the present invention, includes a source private branch change (PBX) 12 that connects a telephone call between a number of user input devices and a public switched telephone network (PSTN) 18. The input devices may be standard telephones 14, a video conferencing system 16 or other types of communication systems such a facsimile machine, etc. Calls from a user input device are typically routed by the PBX 12 on the public switched telephone network 18 to a intended receiver. The intended receiver may be a conventional telephone 22 or a corresponding video conferencing system 24, facsimile machine, etc. The intended receivers are generally coupled to the public switched telephone network 18 through a destination private branch exchange PBX 26.

[0011] As described above, each time a user places a call on the public switched telephone network 18, they are charged for the use of the service. An alternative method of transmitting a telephone call is through the use of a global wide area computer network such as the Internet. To transmit these calls, telephony Internet server 30 is coupled to the PBX 12. The telephony Internet server receives a digitized telephone signal, compresses the signal, and arranges the compressed signal into a series of data packets. An Internet address is added to each packet and the packets are transmitted over the Internet 32 to a receiving telephony Internet server 34, that is coupled to the receiving PBX 26. At the receiving telephony Internet server, the packets are decompressed, combined back into a serial data stream, and supplied to the PBX 26.

[0012] To reduce the cost of communication services,

the communication system of the present invention determines when it is possible to mute a telephone call on the Internet 32 rather than on the PSTN 18. In particular, if a desired recipient's PBX 26 is equipped with a telephony Internet server, and the bandwidth being used by such a server can handle the additional traffic, then a telephone call can be muted on the Internet to avoid paying the additional charges that would be incurred if the call were transmitted on the PSTN 18.

[0013] FIGURE 2 is a flow chart of the steps 50 performed by the PBX 12 shown in FIGURE 1 in order to determine whether a call should be muted on the PSTN 18 or on the Internet 32. Beginning with a step 52, the source PBX receives a telephone number of an intended recipient from a user input device such as a telephone, video conferencing system, facsimile machine, etc. At a step 54, the source PBX determines whether the telephone number received is in the list of recently placed Internet calls. If the answer to step 54 is no, then the PBX 12 begins to place the call on the PSTN 18 using an ISDN or other similar digital format.

[0014] As the call is being set up, the source PBX transmits an information element field that indicates the source PBX has a telephony Internet server with the ability to route the call over the Internet. This information element field is received by the destination PBX and decoded. The destination PBX then responds if it is similarly equipped with a telephony Internet server and if so includes with the response the Internet address of its telephony Internet server. At step 60, the sending PBX determines whether a reply has been received with an address of a destination telephony Internet server within a predetermined time limit. This time limit may be fixed or variable depending on who is attempting to place a call. For example, if the call is placed from an executive phone, then the time limit used before connecting a call on the PSTN may be shorter than the time limit used for calls that originate from the mailroom. If the answer to step 60 is no, then the call set up is completed on the PSTN at a step 62 in a conventional manner.

[0015] If the answer to either the step 54 or 60 is yes, then the PBX forwards the address of the destination telephony Internet server to the telephony Internet server that is coupled to the source PBX at a step 62. The telephony Internet server uses this address to send the packetized telephone call to the intended receiver on the Internet at a step 64. Calls may be immediately rerouted from a PSTN to the Internet upon the determination on that the receiver is equipped with a telephony Internet server. If a response is received indicating that the call can be routed on the Internet after the call has already been set up on the PSTN, then the call can be switched to the Internet after a time period equal to the minimum billing increment on the PSTN. For example, if a call is initially set up on the PSTN and the PSTN bills in one minute increments, the call would be switched to the Internet at the end of the minute

[0016] At any time the quality of the data transmission carried by the telephony Internet servers may degrade such that the call cannot be properly transmitted on the Internet. Therefore, at a step 66, the source PBX queries the telephony Internet server regarding the quality of the call placed on the Internet. Typically, quality is measured by the number of data packets that are transmitted in a given amount of time and the delay introduced by the telephony Internet servers and the network that extends between them to send the packets. Methods for establishing a level of quality on a packetized data network such as the Internet are considered well known to those of ordinary skill in the art and therefore need not be discussed in further detail.

[0017] At a step 68, the sending PBX determines whether the quality of the telephone connection is sufficient to continue the call. If so, processing returns to step 66 until either the quality degrades or the call is finished. If at step 68 it is determined that the quality is insufficient to carry the call, then the source PBX can reroute the telephone call to the intended recipient on the PSTN at a step 70 without user intervention.

[0018] As can be seen from the above description, the present invention is a communication system having alternative paths on which a call can be routed. By determining whether an intended recipient has the ability to transmit data on an alternative path such as the Internet, the alternative path can be used instead of a traditional PSTN. Telephone calls placed on alternative paths avoid the charges that are incurred each time a call is transmitted on the PSTN.

[0019] Although the present invention has been described with respect to the preferred embodiment, it will be appreciated by those skilled in the art that changes can be made. For example, it is possible that the telephony Internet servers could be located at a central telephone office for users that are not connected to the PSTN through a private branch exchange. The central office would query whether a central office that serves the intended recipient is connected with a telephony Internet server and, based on the answer, could route a telephone call either on the Internet or on the PSTN.

Claims

The embodiments of the invention in which an exclusive property or privilege is claimed are defined as follows:

1. A method of muting a telephone call over a public switched telephone network or a global wide area computer network, comprising:

   receiving a telephone number that is associated with a desired recipient of the telephone call;

   setting up a telephone call to the desired recip-

ient on the public switched telephone network;
determining whether the recipient has a capability to receive the telephone call on the global wide area computer network before the telephone call is connected a the public switched telephone network; and
connecting the telephone call on the global wide area computer network if the recipient has the capability to receive the telephone call on the global wide area network.

2. The method of Claim 1, wherein the telephone call is transmitted on an ISDN line over the public switched telephone network and wherein the step of determining whether the recipient has the capability to receive the telephone on the global wide area computer network comprises:

transmitting a signaling packet to the desired recipient that indicates that a source of the telephone call is equipped with a telephony Internet server, and
receiving from the desired recipient a signaling packet that indicates that the desired recipient is equipped with a telephony internet server.

3. The method of Claim 2, further comprises:

monitoring a time period before a signaling packet is received from the desired recipient that indicates that the desired recipient is equipped with a telephony internet server; and comparing the time period with a predetermined time threshold and if the time period exceeds the time threshold, then completing the telephone call set up on the public switched telephone network.

4. The method of Claim 3, wherein the time threshold is varied depending on the source of the telephone call.

5. The method of Claim 3, further comprising:

determining if a signaling packet is received from the desired recipient that indicates the desired recipient is equipped with a telephony Internet server after the telephone call has been set up on the public switched telephone network, and if so, connecting the telephone call on the global wide area computer network at the and of a minimum billing increment of the public switched telephone network.

6. The method of Claim 1, further comprising monitoring the quality of the telephone call on the global wide area computer network; and
transfering the telephone call to the public switched telephone network if the quality is degraded.

7. The method of Claim 6, wherein the telephone call transmitted as a number of data packets on the global wide area network and wherein the step of monitoring the quality of the telephone call on the global wide area network, comprises:

determining the rate at which data packets are sent by and received by the telephony Internet servers; and
determining if the rate of the data packets is less than a rate threshold and if so, declaring the quality of the telephone call to be degraded.

8. The method of Claim 6, wherein the telephone call is transmitted as a number of data packets on the global wide area network and wherein the step of monitoring the quality of the telephone call on the global wide area network comprises:

determining a delay time required before a data packet is transmitted on the global wide area network; and
determining if the delay time is greater than a delay time threshold and if so declaring the quality of the telephone call to be degraded.

9. The method of Claim 6, wherein the telephone call is transmitted as a number of data packets on the global wide area network and wherein the step of monitoring the quality of the telephone call on the global wide area network comprises:

determining a rate at which data packets are transmitted and received by the telephony Internet servers and a delay time required before a data packet is transmitted on the global wide area network;
comparing the rate and delay time to a rate and delay time threshold and if the rate is less than the rate threshold, or if the delay time is greater than the delay threshold, declaring the quality of the telephone call to be degraded.

10. A communication system for muting a telephone call over a public switched telephone network or over a global wide area computer network, comprising:

a private branch exchange that connects calls between one or more source telephones and the public switched telephone network;
a telephony Internet server, coupled to the private branch exchange that can route a telephone call over the global wide area network, wherein the private branch exchange operates

to place a call to a second private branch exchange that connects the telephone call to an intended recipient network and to determine whether the second private branch exchange is coupled to a telephony Internet server and if so, routes the telephone call to the intended recipient on the global wide area computer network.

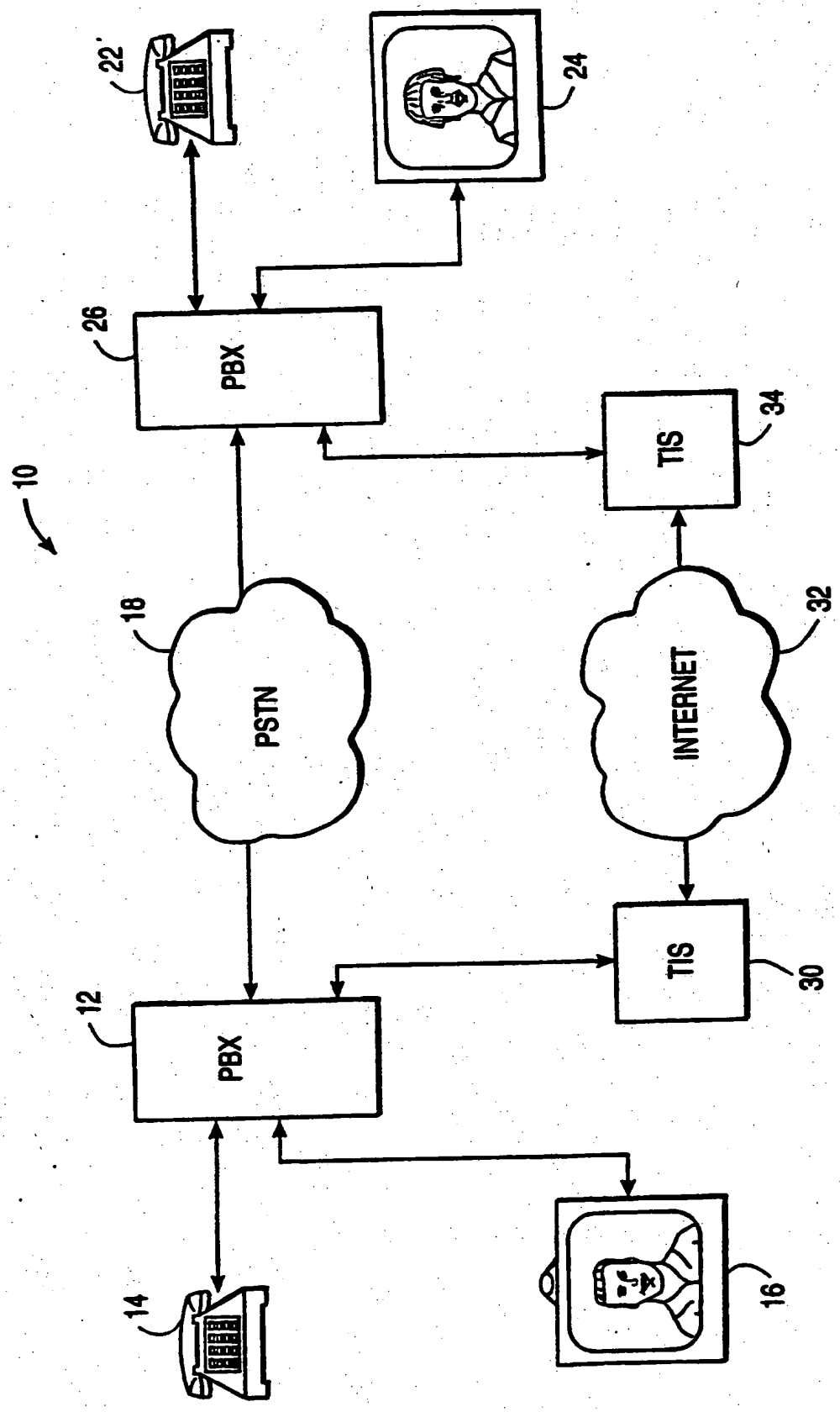

FIG_1

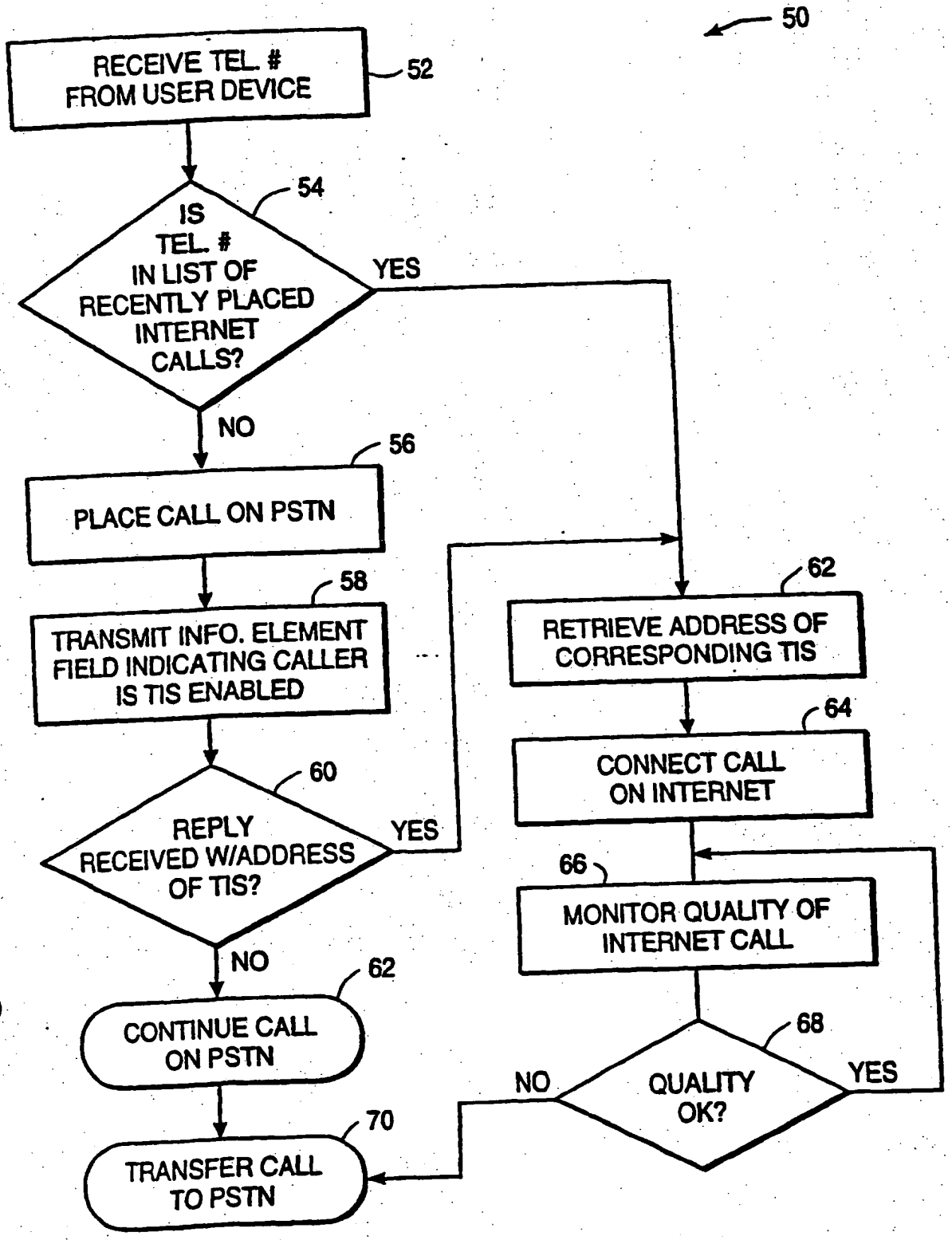
50
RECEIVE TEL. # FROM USER DEVICE
52
IS TEL. # IN LIST OF RECENTLY PLACED INTERNET CALLS?
54
YES
NO
PLACE CALL ON PSTN
56
TRANSMIT INFO. ELEMENT FIELD INDICATING CALLER IS TIS ENABLED
58
REPLY RECEIVED W/ADDRESS OF TIS?
60
YES
NO
CONTINUE CALL ON PSTN
62
TRANSFER CALL TO PSTN
70
RETRIEVE ADDRESS OF CORRESPONDING TIS
62
CONNECT CALL ON INTERNET
64
MONITOR QUALITY OF INTERNET CALL
66
QUALITY OK?
68
NO
YES

FIG_2

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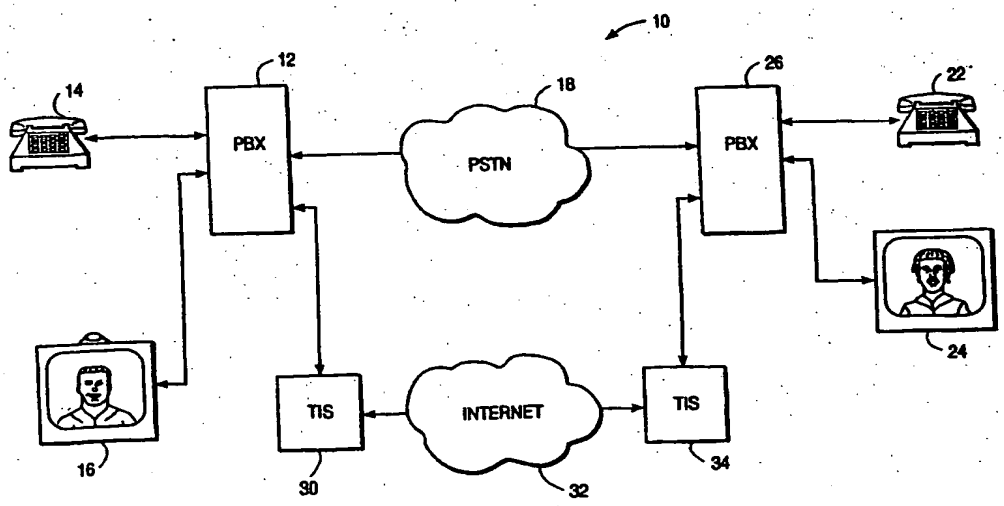

FIG_1

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EUROPEAN SEARCH REPORT

Application Number
EP 98 11 8543

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
P,X	WO 98 11704 A (DIALNET INC) 19 March 1998 (1998-03-19) * page 12, line 24 - page 13, line 24 *	1	H04M7/00 H04M7/12 H04Q3/62 H04L29/06
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A	HOUGHTON T F ET AL: "A PACKET TELEPHONY GATEWAY FOR PUBLIC NETWORK OPERATORS" ISS '97. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM). GLOBAL NETWORK EVOLUTION: CONVERGENCE OR COLLISION? TORONTO, SEPT. 21 - 26, 1997, ISS. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM), TORONTO, P, vol. 2, 21 September 1997 (1997-09-21), pages 35-44, XP000704453		
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The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	17 June 2002	Vandevenne, M

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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The members are as contained in the European Patent Office EDP file on
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